WHAT IS CLAIMED IS:

1. A method for approximating $y(n)=1/x(n)$ in FM demodulation, where $x(n)= I^2(n)+Q^2(n)$, comprising:

(a) receiving a prior estimated value of $1/x(n)$;

(b) receiving a present value of $x(n)$;

(c) adjusting the prior estimated value of $1/x(n)$ to compensate for an error between the prior estimated value of $1/x(n)$ and the present value of $1/x(n)$; and

(d) outputting the adjusted prior estimated value of $1/x(n)$ as the present value of $1/x(n)$.

2. The method of claim 1, wherein the prior estimated value of $1/x(n-1)$ equals $1/(I^2(n-1)+Q^2(n-1))$, wherein $I(n)$ is an input signal and $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$.

3. The method of claim 2, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

4. The method of claim 1, wherein the present value $x(n)$ equals $I^2(n)+Q^2(n)$, and wherein $I(n)$ is an input signal and $Q(n)$ is quadrature-phase signal of $I(n)$.

5. The method of claim 4, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

6. The method of claim 1, wherein an error signal equals $(1-x(n)y(n-1))a$, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1)= 1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and "a" is a scaling coefficient.

7. The method of claim 6, wherein the input signal I(n) comprises a band pass filtered secondary audio program signal.

8. The method of claim 1, wherein the Y(n) signal equals y(n-1) + (1-x(n)(y(n-1))a, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1)= 1/(I^2(n-1)+Q^2(n-1))$, I(n) is an input signal, Q(n) is a quadrature-phase signal of the input signal I(n), and "a" is a scaling coefficient.

9. The method of claim 8, wherein the input signal I(n) comprises a band pass filtered secondary audio signal.

10. A method for demodulating an FM signal FM(n) from a secondary audio program signal, comprising:

(a) receiving in-phase I(n) and quadrature-phase Q(n) portions of the FM(n) signal

(b) generating a first portion of the FM(n) signal that is equal to $I(n)Q'(n)-I'(n)Q(n)$;

(c) determining a value z(n) based on the first portion of the FM(n) signal;

(d) generating a second portion of the FM(n) signal that is equal to $1/ I^2(n)+Q^2(n)$, wherein $I^2(n)+Q^2(n)$ is equal to x(n) and y(n)=1/x(n);

(e) generating a value for y(n) based on 1/x(n) that equals y(n-1) + (1-x(n)y(n-1))a; and

(f) multiplying the z(n) value and the y(n) value to produce the FM(n) signal.

11. A system for approximating $y(n)=1/x(n)$ in FM demodulation, where $x(n)= I^2(n)+Q^2(n)$, comprising:

means for receiving a prior estimated value of $1/x(n)$;

means for receiving a present value of $x(n)$;

means for adjusting the prior estimated value of $1/x(n)$ to compensate for an error between the prior estimated value of $1/x(n)$ and the present value of $1/x(n)$; and

means for outputting the adjusted prior estimated value of $1/x(n)$ as the present value of $1/x(n)$.

12. The system of claim 11, wherein the prior estimated value of $1/x(n-1)$ equals $1/(I^2(n-1)+Q^2(n-1))$, wherein $I(n)$ is an input signal and $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$.

13. The system of claim 12, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

14. The system of claim 11, wherein the present value $x(n)$ equals $I^2(n)+Q^2(n)$, and wherein $I(n)$ is an input signal and $Q(n)$ is quadrature-phase signal of $I(n)$.

15. The system of claim 14, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

16. The system of claim 11, wherein an error signal equals $(1-x(n)y(n-1))a$, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1)= 1/(I^2(n-1)+Q^2(n-1))$, $I(n)$ is an input signal, $Q(n)$ is a quadrature-phase signal of the input signal $I(n)$, and "a" is a scaling coefficient.

17. The system of claim 16, wherein the input signal $I(n)$ comprises a band pass filtered secondary audio program signal.

18. The system of claim 11, wherein the Y(n) signal equals y(n-1) + (1-x(n)(y(n-1))a, wherein $x(n) = I^2(n)+Q^2(n)$, $y(n-1)= 1/(I^2(n-1)+Q^2(n-1))$, I(n) is an input signal, Q(n) is a quadrature-phase signal of the input signal I(n), and "a" is a scaling coefficient.

19. The system of claim 18, wherein the input signal I(n) comprises a band pass filtered secondary audio signal.

20. A method for approximating y(n)=1/x(n) in FM demodulation, where $x(n)= I^2(n)+Q^2(n)$, comprising:

(a) receiving 1/x(n-1);

(b) receiving x(n);

(c) adjusting 1/x(n-1) to compensate for an error between 1/x(n-1) and 1/x(n); and

(d) outputting the adjusted 1/x(n-1) as 1/x(n).